

## The Audio- Is of the Main Components of Multimedia Technologies

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**Abstract:** The article deals with one of the main components of multimedia technologies - audio and its properties, the creation of audio effects and its use in multimedia applications.

**Keywords:** multimedia, animation, graphics, sound, microphone, audio systems, optical CD, TV, slide, presentation.

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### INTRODUCTION

In today's world, millions of people listen to thousands of audio recordings every day. Today we can listen to an audio recording of some event taking place in another part of the world, music tracks of our favorite bands, without attending concerts. And all this is possible thanks to information technologies for processing video and audio information. Audio information is considered one of the components of multimedia technologies. Audio information is actively used in teaching.

Informatization has largely transformed the process of obtaining knowledge. New learning technologies based on information and communication make the educational process more intense, increase the speed of perception, understanding and, importantly, the depth of assimilation of a large amount of knowledge. The concept of information technology of education characterizes the process of preparing and transmitting information to the student. The means of implementing this process are computer hardware and software[2].

To develop lessons using a computer, the teacher must know the functionality and conditions for using each of these components. Both technical and software tools have their own specifics and in a certain way affect the educational process. Here are some methodological possibilities of information technology tools:

- access to a large amount of information presented in an entertaining way, through the use of multimedia;
- development of skills to process information when working with computer catalogs and directories;
- strengthening learning motivation (through games, multimedia);

Information technology training involves the use of specialized software along with computer technology. A software tool for educational purposes is a software tool in which a certain subject area is recreated, where the technology of its study is implemented, conditions are created for the implementation of various types of educational activities. Such software tools that functionally support various types of the educational process are called pedagogical software tools. Currently, there are a large number of different classifications and typologies of pedagogical software[3].

### RESULTS and DISCUSSIONS

Sound information (audio information) in multimedia systems is usually technologically

represented as an audio series, that is, a sequence of sound pressure amplitude values recorded in digital form. There are two main types of sound files: digitized audio and notated music.

Sound is a wave with a continuously changing amplitude and frequency. The greater the amplitude of the source oscillations, the greater the intensity of the sound waves excited by it, therefore, the louder the sound perceived by our hearing organs. The intensity of sound in engineering is usually measured in decibels, abbreviated as dB (dB). The frequency of sound is measured in units per second, or, in other words, in hertz, abbreviated as Hz. Frequency determines the pitch that our ear detects. Small (low) frequencies are associated in our minds with dull bass, and large (high) frequencies are associated with a piercing whistle. A person is able to perceive sounds in the frequency range from about 20 to 20,000 Hz, lower and higher than the threshold of hearing, the frequencies are called infrasound and ultrasound, respectively. To encode sound, it is necessary to measure the amplitude of the signal at certain intervals. The procedure is called sound sampling, and the frequency with which the signal amplitude is determined is the sampling frequency[4].

The amplitude of the signal, determined at each moment of time, must also be presented in numerical form. Acceptable sound quality is obtained when using 16 bits (2 bytes). This bit depth is used most often. To record stereo sound, two independent audio channels should be encoded simultaneously with the above parameters. Up to 8 channels can be required to create the popular "surround sound". To store sound files, a large number of formats have been developed that use various compression algorithms. Uncompressed audio in Windows is stored in the Microsoft Wave (Windows PCM, WAV) format. Formats that support compression, as in the case of graphic information, are divided into two groups: using lossless and lossy compression. The formats of the first group (the most famous are FLAC, Monkey's Audio, WavPack, TTA) compress sound. In the practice of video editing, audio formats are used that compress sound with loss. The most common compressed audio format is MPEG-1 Layer 3(MP3). Other lossy audio formats are currently being actively developed: Ogg Vorbis (OGG), Windows Media Audio (WMA), MPEG-4 AAC (MP4), which provide better sound quality than MP3 at the same file size.

The process of converting an analog signal into a form compatible with a digital communication system begins with a sampling process. To convert an analog signal into a discrete signal, in most cases, the instantaneous value of the signal is sampled at strictly defined time intervals. The result of this sampling process is a pulse-amplitude modulation (PAM) signal. Note that it is easy to recover an analog signal with a certain degree of accuracy by applying a simple low-pass smoothing filter to the PAM.

Established data compression techniques such as RLE can be used to compress audio files without loss, but the result is highly dependent on the specific audio data. Some sounds will compress well with RLE, but poorly with statistical algorithms. Other sounds are more suited to statistical compression. It is possible to achieve better results when compressing audio with loss of some of the audio information by developing compression methods that take into account the peculiarities of sound perception. They remove that part of the data that remains inaudible to the hearing organs. The existence of an audibility threshold provides a basis for constructing lossy audio compression methods. You can delete all samples whose value lies below this threshold. In addition, two more properties of the human hearing organs are used to effectively compress sound. These properties are called frequency masking and time masking[4].

There are many different ways to store digital audio. As we said, digitized sound is a set of signal amplitude values taken at certain time intervals. Thus, firstly, a block of digitized audio information can be written to a file "as is", that is, by a sequence of numbers (amplitude values). In this case, there are two ways to store information.

PCM (Pulse Code Modulation - pulse code modulation) - a method of digital encoding of a signal by recording the absolute values of the amplitudes (there are signed or unsigned representations). It is in this form that data is recorded on all audio CD.

ADPCM (Adaptive Delta PCM - adaptive relative pulse-code modulation) - recording signal values not in absolute, but in relative amplitude changes (increments). Secondly, you can compress or simplify the data so that it takes up less memory than if it were written "as is".

Sound processing should be understood as various transformations of sound information in order to change some sound characteristics. Sound processing includes methods for creating various sound effects, filtering, as well as methods for cleaning sound from unwanted noise, changing timbre, etc. All this huge set of transformations comes down, ultimately, to the following main types:

- Amplitude transformations;
- Frequency conversions;
- Phase transformations;
- Temporary transformations;

A discussion of each of these types of transformations can become a whole scientific work. It is worth giving some practical examples of using these types of transformations when creating real sound effects:

- Echo (echo);
- Reverberation (repetition, reflection);
- Chorus (chorus);

Of course, as in all other areas, signal processing also has problems that are a kind of stumbling block. So, for example, when decomposing signals into a frequency spectrum, there is an uncertainty principle that cannot be overcome. The principle says that it is impossible to obtain an accurate spectral picture of a signal at a specific point in time: either to obtain a more accurate spectral picture, you need to analyze a larger time section of the signal, or, if we are more interested in the time when this or that change in the spectrum occurred, you need to sacrifice the accuracy of the spectrum itself. In other words, it is impossible to obtain the exact spectrum of the signal at a point - the exact spectrum for a large section of the signal, or a very approximate spectrum, but for a short section[2].

The spectrum of sound processing software is very wide. The most important class of software is digital audio editors. The main features of such programs are, at a minimum, providing the ability to record (digitize) audio and save it to disk. Developed representatives of this kind of programs allow much more: recording, multi-channel audio mixing on several virtual tracks, processing with special effects (both built-in and externally connected - more on that later), noise removal, have advanced navigation and tools in the form of a spectroscope and others virtual instruments, control / manageability of external devices, audio conversion from format to format, signal generation, recording to CDs and much more. Some of these programs are Cool Edit Pro (Syntrillium), Sound Forge (Sonic Foundry), Nuendo (Steinberg), Samplitude Producer (Magix), Wavelab (Steinberg).

A group of programs that is no less important in a functional sense is sequencers (programs for writing music). Most often, such programs use a MIDI synthesizer (external hardware or built into almost any sound card, or software, organized by special software). Bright representatives of this class of programs: Cubase (Steinberg), Logic Audio (Emagic), Cakewalk (Twelve Tone Systems) and the already mentioned Finale.

Many programs belonging to a particular group of software allow you to connect the so-called "plug-ins" - external plug-ins that expand the possibilities of sound processing. This became possible as a result of the emergence of several standards for the interface between the program and the plug-in. To date, there are two main interface standards: DX and VST. The existence of standards allows you to connect the same plug-in to completely different programs without worrying about conflicts and malfunctions. Speaking about the plugins themselves, it must be

said that this is just a huge family of programs. Usually, one plug-in is a mechanism that implements a specific effect, such as a reverb or a low-pass filter. Among the interesting plug-ins, we can recall, for example, iZotope Vinyl - it allows you to give the sound the effect of a vinyl record (see the screenshot - an example of the plug-in working window in the Cool Edit Pro environment), Antares AutoTune allows you to correct the sound of vocals in semi-automatic mode, and Orange Vocoder is a wonderful vocoder (a mechanism for making various instruments sound like the sound of a human voice)[4].

Audio analyzer programs specially designed for measuring audio data analysis. Such programs help to present audio data in a more convenient way than conventional editors, as well as to carefully study them using various tools, such as FFT analyzers (builders of dynamic and static frequency response), sonogram builders, and others. One of the most famous and developed programs of this kind is the Spectra LAB program (Sound Technology Inc.), slightly simpler, but powerful - Analyzer2000 and Spectrogram.

Specialized audio restorers also play an important role in sound processing. Such programs allow you to restore the lost sound quality of audio material, remove unwanted clicks, noises, crackles, specific noise from audio cassette recordings, and perform other audio adjustments. Programs of this kind: Dart, Clean (from Steinberg Inc.), Audio Cleaning Lab. (from Magix Ent.), Wave Corrector[4].

Trackers. Trackers are a separate category of sound programs designed specifically for creating music. Tracker modules (files created in trackers are usually called modules), as well as MIDI files, contain a score in accordance with which the instruments should be played. In addition, they contain information about what effects and at what point in time should be applied when playing a particular instrument. The main tracker programs are Scream Tracker, Fast Tracker, Impulse Tracker, OctaMED SoundStudio, MAD Tracker, ModPlug Tracker.

It should be mentioned that there is a huge amount of other audio software: audio players (the most prominent: WinAMP, Sonique, Apollo, XMPlay, Cubic Player), plug-ins for players (among the "enhancers" of audio sound - DFX, Enhancer, iZotope Ozone), utilities for copying information from an audio CD (ExactAudioCopy, CDex, AudioGrabber), audio stream interceptors (Total Recorder, AudioTools), audio encoders (MP3 encoders: Lame encoder, Blade Encoder, Go-Go and others; VQF encoders: TwinVQ encoder, Yamaha SoundVQ, NTT TwinVQ; AAC encoders: FAAC, PsyTel AAC, Quartex AAC), audio converters (for converting audio information from one format to another), speech generators and many other specific and general utilities. Of course, all of the above is only a small fraction of what can be useful when working with sound[2].

The global Internet is a human-made communication environment. The term "communication medium" (data transmission medium, information) is currently used in several meanings. Recommendations for sound transmission systems over the Internet have been developed, consisting in the use of:

- modern compression algorithms that allow achieving acceptable sound quality at high compression ratios;
- a request-response exchange mode, with retransmissions upon detection of a loss;
- an accumulation buffer at the information receiver, the size of which, in terms of the duration of playback of the corresponding sound fragment, must be not less than the maximum packet delay in the network.

Audio is actively used in the global network. In Web sites (multimedia projects), sound can be the main content of the site, but it can also perform a service or aesthetic function. A service function involves informing the user about events that occur, warning or supporting the visitor's actions. Sound as an aesthetic component, such as musical accompaniment, is needed to form an emotional background that is favorable for the assimilation of information presented on a Web page.

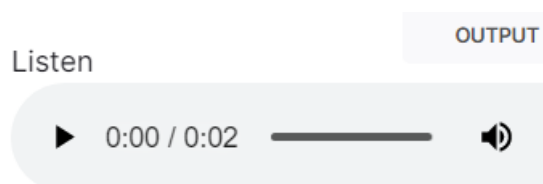
Unfortunately, at present, the speed of data transfer over the global network (under the conditions of a conventional modem connection) and the existing methods of compressing audio information do not allow creating full-sounding sites. In addition, experience shows that most users will not wait for the download to complete if it takes too long [2]. One of the main reasons a user leaves a site is slow loading. This is due to the fact that sound files can overload the page. Therefore, it is necessary to accurately determine the file formats, the degree of compression, the scope and use the sound where it really helps the user or causes him a positive reaction.

There are many different methods for displaying audio information in web pages. For example, using the HTML `<audio>` tag. The `<audio>` HTML element is used to embed audio content into a document. It can contain one or more audio sources, represented by either the `src` attribute or the `<source>` element - the browser will pick the one most appropriate. It can also be used for streaming media using the `MediaStream` interface.

HTML code:

```
<figure>
<figcaption>Listen:</figcaption>
<audio
controls
src="/media/audio/mytext.mp3">
Your browser does not support the <code>audio</code> element.
</audio>
</figure>
```

Result:



The example above shows a simple use of the `<audio>` element. We can add other attributes to specify details such as autoplay and repeat, whether we want to display standard browser audio controls, and so on.

The content between the opening and closing tags of the `<audio>` element (in the example above) is rendered in browsers that do not support the element.

Not all browsers support the same audio formats. You can provide multiple sources inside nested `<source>` elements and then the browser will use the first one it understands:

```
<audio controls>
<source src="MySound.mp3" type="audio/mpeg">
<source src="MySound.ogg" type="audio/ogg">
<p>Your browser does not support HTML5 audio. Replacement
<a href="MySound.mp4">audio link</a></p>
</audio>
```

Thus, the use of sound in one form or another in some cases can make the Internet project more attractive to the user. Sometimes its use is directly related to the content of the project. But a large amount of sound information and, as a result, large volumes of sound files do not allow

creating a full-fledged sound accompaniment [3].

## CONCLUSIONS

The most commonly used IT tools in the educational process include electronic textbooks and manuals demonstrated using a computer and a multimedia projector, Internet educational resources, video and audio equipment, electronic encyclopedias and reference books, DVDs and CDs with pictures and illustrations, simulators and testing programs, research work and projects. From the point of view of orientation, IT tools can be classified as follows: for information retrieval, for working with texts, for automatic translation, for storing and accumulating information, for processing and reproducing graphics and sound, as well as for communication. The formation of listening skills is one way to solve this problem. Many sites provide audio information. For example, the Internet is also useful in studying the grammar and vocabulary of foreign languages. Learning will be more effective if new knowledge is perceived by multiple senses, such as sight and hearing. Audio recordings, visual aids, videos contribute to a better assimilation of knowledge. Also, the discussion method of teaching helps to better consolidate the acquired knowledge, apply it and attract the attention of the student.

Today, with the help of information technology, the learning process can be closer to real conditions than ever before. So, computers are able, for example, to perceive this or that information, process it, store certain data in memory, play audio and video materials, etc. Along with the listed capabilities, computers significantly expand the teacher's opportunities in terms of individualizing learning and enhancing cognitive activity. students in the learning process. Thus, information technology tools make it possible to build the learning process in accordance with the individual needs of students, which makes it possible to work at their own pace, independently choosing the speed and amount of learning that they consider optimal for themselves.

Modern methods of audio data processing are not without drawbacks and therefore can be improved. The article showed the main problems of common audio processing and encoding algorithms. An analysis was also made of software for processing audio information, their ability to improve existing audio information technologies. Sound information is one of the most promising technologies in the field of digital signal processing in the global Internet. As a result, the most common audio formats and software for processing sound information were proposed, they learned about the role of audio in the educational process.

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